WHAT IS CLAIMED IS:

1. A method of generating a wide-band speech signal from a first narrow-band speech signal, the method comprising:

analyzing the first narrow-band speech signal to generate one or more parameters;

synthesizing a lower frequency-band signal based on at least one of the one or more parameters; and

combining the synthesized lower frequency-band signal with a second narrow-band speech signal that is derived from the first narrow-band speech signal,

wherein:

the one or more parameters include a pitch frequency parameter; and

synthesizing the lower frequency-band signal based on at least one of the one or more parameters comprises generating continuous sine tones that are based on the pitch frequency parameter.

2. The method of claim 1, further comprising generating the second narrow-band speech signal by a technique that includes up-sampling the narrow-band speech signal.

3. The method of claim 1, wherein the second narrow-band speech signal is the first narrow-band speech signal.

4. The method of claim 1, wherein:

the narrow-band speech signal comprises a plurality of narrow-band speech signal segments;

the pitch frequency parameter is estimated for each of the narrow-band speech signal segments; and

the continuous sine tones are changed gradually during a first part of each speech signal segment.

5. The method of claim 4, wherein synthesizing the lower frequency-band signal based on at least one of the one or more parameters further comprises adaptively changing an amplitude level of the continuous sine tones based on an amplitude level of at least one formant in the narrow-band speech signal segment.

6. The method of claim 5, wherein the at least one formant in the narrow-band speech signal segment is a first formant in the narrow-band speech signal segment.

7. The method of claim 5, wherein adaptively changing the amplitude level of the continuous sine tones based on the amplitude level of at least one formant in the narrow-band speech signal segment comprises:

adaptively changing an amplitude level of the continuous sine tones by an amount, $g_l(m)$, given by:

$$g_l(m) = C_l \cdot \sqrt{\frac{\sum_{l=0}^{p} a(l) \cdot \gamma_{xx}(l)}{\left| \sum_{l=0}^{p} a(l) \cdot e^{-j2\pi l f_{Nl}} \right|^2}} \quad ,$$

where C_l is a constant; m is a segment number; γ_{xx} is an autocorrelation value of the narrow-band speech signal, x; f_{Nl} is a frequency of a first formant of the narrow-band speech signal; and p is an order of a linear prediction filter.

8. The method of claim 5, wherein the continuous sine tones, $s(n)$, are generated in accordance with:

$$s(n) = \sum_{i=1}^{N} s_i(n),$$

where the summation range $i = 1$ to N is selected such that all sine tones will be added together, and:

$$s_i(n) = \begin{cases} \left(g_i(m-1) + n\frac{g_i(m)-g_i(m-1)}{L_l}\right) \sin\left(i(\phi(m)+n)\left(\omega(m-1)+n\frac{\omega(m)-\omega(m-1)}{L_l}\right)\right), & n = 0,\ldots,L_l \\ g_i(m)\sin(i(\phi(m)+n)\omega(m)), & n = L_l+1,\ldots,L-1 \end{cases}$$

where $\phi(m)$ is a phase compensation needed to maintain a continuous sinusoid within segments, $\omega(m)$ is the pitch frequency of a current speech signal segment m, L is the number of samples in each speech signal segment, and L_l is the end sample of the soft transition within each speech signal segment.

9. The method of claim 1, wherein synthesizing the lower frequency-band signal based on at least one of the one or more parameters further comprises lowpass filtering the continuous sine tones.

10. The method of claim 9, wherein lowpass filtering the continuous sine tones is performed with an upper cutoff frequency substantially equal to 300 Hz.

11. An apparatus for generating a wide-band speech signal from a first narrow-band speech signal, the apparatus comprising:

logic that analyzes the first narrow-band speech signal to generate one or more parameters;

logic that synthesizes a lower frequency-band signal based on at least one of the one or more parameters; and

logic that combines the synthesized lower frequency-band signal with a second narrow-band speech signal that is derived from the first narrow-band speech signal,

wherein:

5   the one or more parameters include a pitch frequency parameter; and

the logic that synthesizes the lower frequency-band signal based on at least one of the one or more parameters comprises logic that generates continuous sine tones that are based on the pitch frequency parameter.

12.   The apparatus of claim 11, further comprising logic that generates the

10   second narrow-band speech signal by a technique that includes up-sampling the narrow-band speech signal.

13.   The apparatus of claim 11, wherein the second narrow-band speech signal is the first narrow-band speech signal.

14.   The apparatus of claim 11, wherein:

15   the narrow-band speech signal comprises a plurality of narrow-band speech signal segments;

the pitch frequency parameter is estimated for each of the narrow-band speech signal segments; and

the continuous sine tones are changed gradually during a first part of each

20   speech signal segment.

15.   The apparatus of claim 14, wherein the logic that synthesizes the lower frequency-band signal based on at least one of the one or more parameters further comprises logic that adaptively changes an amplitude level of the continuous sine tones based on an amplitude level of at least one formant in the narrow-band

25   speech signal segment.

16.   The apparatus of claim 15, wherein the at least one formant in the narrow-band speech signal segment is a first formant in the narrow-band speech signal segment.

17.   The apparatus of claim 15, wherein the logic that adaptively changes the amplitude level of the continuous sine tones based on the amplitude level of at least one formant in the narrow-band speech signal segment comprises:

logic that adaptively changes an amplitude level of the continuous sine tones by an amount, $g_i(m)$, given by:

$$g_i(m) = C_i \cdot \sqrt{\frac{\sum_{l=0}^{p} a(l) \cdot \gamma_{xx}(l)}{|\sum_{l=0}^{p} a(l) \cdot e^{-j2\pi l f_{Nl}}|^2}} \quad ,$$

where C_i is a constant; m is a segment number; γ_{xx} is an autocorrelation value of the narrow-band speech signal, x; f_{Nl} is a frequency of a first formant of the narrow-band speech signal; and p is an order of a linear prediction filter.

18.   The apparatus of claim 15, wherein the continuous sine tones, $s(n)$, are generated in accordance with:

$$s(n) = \sum_{i=1}^{N} s_i(n),$$

where the summation range $i = 1$ to N is selected such that all sine tones will be added together, and:

$$s_i(n) =$$
$$\begin{cases} \left(g_l(m-1) + n\frac{g_l(m)-g_l(m-1)}{L_l}\right)\sin\left(i(\phi(m)+n)\left(\omega(m-1)+n\frac{\omega(m)-\omega(m-1)}{L_l}\right)\right), & n = 0,\ldots,L_l \\ g_l(m)\sin(i(\phi(m)+n)\omega(m)), & n = L_l+1,\ldots,L-1 \end{cases}$$

where $\phi(m)$ is a phase compensation needed to maintain a continuous sinusoid within segments, $\omega(m)$ is the pitch frequency of a current speech signal segment m, L is the number of samples in each speech signal segment, and L_l is the end sample of the soft transition within each speech signal segment.

5   19.   The apparatus of claim 11, wherein the logic that synthesizes the lower frequency-band signal based on at least one of the one or more parameters further comprises a lowpass filter that lowpass filters the continuous sine tones.

20.   The apparatus of claim 19, wherein the lowpass filter has an upper cutoff frequency substantially equal to 300 Hz.